# Transport of Video over Partial Order Connections

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Abstract. *A Partial Order and partial reliable Connection (POC) is an end-to-end transport connection authorized to deliver objects in an order that can differ from the transmitted one. Such a connection is also authorized to lose some objects. The POC concept is motivated by the fact that heterogeneous best-effort networks such as Internet are plagued by unordered delivery of packets and losses, which tax the performances of current applications and protocols. It has been shown, in several research works, that out of order delivery is able to alleviate (with respect to CO service) the use of end systems' communication resources. In this paper, the efficiency of out-of-sequence delivery on MPEG video streams processing is studied. Firstly, the transport constraints (in terms of order and reliability) that can be relaxed by MPEG video decoders, for improving video transport, are detailed. Then, we analyze the performance gain induced by this approach in terms of blocking times and recovered errors. We demonstrate that POC connections fill not only the conceptual gap between TCP and UDP but also provide real performance improvements for the transport of multimedia streams such MPEG video.*
Keywords: *MPEG, Transport Protocol, Video Transport, Video Decoder, QoS mapping.*

## 1 Introduction

The increasing capabilities of high performance end systems and communication networks have greatly accelerated the development of distributed computing. The trend towards global distributed computing highlights that currently used Transport layers are not able to offer an efficient support to distributed multimedia applications. Two main families of solutions have been proposed to overcome these deficiencies: either to improve hardware platforms and software implementations or to develop new mechanisms that are better adapted to new applications needs.

The first family of solutions (i.e. the improvement of traditional protocols such as TCP/IP) considers that new protocols may add functionality (for certain applications) but their services could not surpass the services already provided by traditional approaches. This tendency is also encouraged by the performance gain that may be achieved in applying mechanisms such as better hardware interfaces [31][35] or better flow and error control mechanisms [24].

The second approach consists in creating new mechanisms that are more conformant with new applications profiles [11][14]. For achieving this goal, it is necessary to tailor the communication system functionality to the application's needs. The Partial Order and Reliability Connection (POC) concept [1][12][14] conforms to this approach.

The partial order and the partial reliability concepts are the generalization (in terms of order and reliability) of the services provided by a protocol. The different possibilities of order and reliability form a space of protocol services. A point in this space corresponds to a particular service defined by a couple of values of order and reliability (see Figure 1 ). Thus, traditional protocols such as TCP and UDP become particular cases of this generalization.

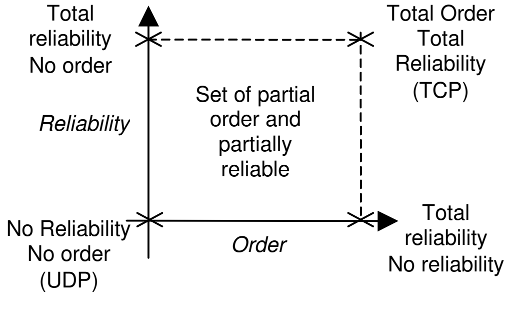

**Figure 1. Representation of the order and reliability space.**

In other words, a partial order and partial reliability service is an end-to-end connection where (1) data delivery order can differ from the transmission one and (2) some losses are authorized. It has been shown that in a best-effort environment this new approach improves speed-up transfer and resources saving at both sender and receiver sides [16][18][23].

The goal of this paper is to analyze the effects of a POC approach on a MPEG video transport system. This goal is reached in two steps. First, we define a new transport service that allows one to maintain playback continuity as well as to achieve efficient error recovery for MPEG streams is. Then, we demonstrate the efficiency of this new transport service by implementing and using a MPEG decoder that is able to process information in a not orderly fashion.

The paper is organized as follows. Section 2 presents an overview of video transport issues. Section 3 describes the MPEG streams' features that allow their order and reliability constraints to be relaxed for their transport. Based on these considerations, Section 4 proposes a new transport service for MPEG video. Section 5 analyzes the performance gain, and section 6 presents the implementation of the transport service. Finally, section 7 gives concluding remarks.

## 2 Video and Transport Protocols Today

Two main considerations have motivated this work: on the one hand, the inadequacy of traditional protocols to transport real time video and, on the other hand, the unexplored possibilities of order management as a mean for improving a transport protocol.

### 2.1 Limitations of traditional protocols

Traditional protocols have not been created with multimedia applications features in mind. Among multimedia applications, those managing video streams entail the most acute new features: continuous synchronization and temporal constraints. For supporting such applications, current protocols offer only two classes of basic services: Connection Oriented (CO) or Connection Less (CL). The use of such basic services for video transport involves several drawbacks. Indeed, CO services ensure reliability at the expense of latency increase. Moreover the incompatibility of video applications and CO services is strengthened by current best-effort global distributed systems such the Internet. Conversely, a protocol providing a CL service does not ensure reliability and, as it does not implement retransmission recovery, a CL protocol ensures a minimal number of blocking times. Apparently, such protocols provide a more adequate service for video transport since video requires continuity of delivery as well as it supports losses. However, some degree of reliability is required to ensure an acceptable quality of video presentation. Indeed, when compression algorithm are based on predictive or hierarchical coding techniques, the total or partial loss of a given frame can entail the loss of one or several pictures depending on it. Thus, considering that CL service are not fully adequate for video transport, a video distributed application based on a CL service must implement some transport functionality, which may represent an important implementation overhead. Examples of such an approach are the INRIA's IVS teleconferencing system [37], and the vic systems widely used on the Mbone [25]. Moreover, in order to reduce this kind of implementation effort, a compiler tool has been proposed in [5] and an automated generation of dynamically adaptable protocols is defined in [33][34].

In order to provide an optimal service to video applications, a transport protocol must consider the particularities of the application. The error recovery mechanism of a protocol is an essential aspect. Ideally, an error recovery mechanism adapted to video transport should not disrupt the continuous delivery of the media. It will be shown in the following that the ability of video for supporting partial losses and disordering can significantly help to avoid risks of delivery disrupt. The amount of losses a video stream can support depends essentially on the presentation quality the final receiver is ready to accept. In order to determine which video elements can be lost or received in disorder, the error recovery mechanism must know the semantic structure of the bit-stream associated to the media (which depends on the coding algorithm). Current protocols are neither able to adapt their services to such a bit-stream's semantic nor able to tradeoff between time and network errors in order to obtain an optimal service from the network.

In summary, there is nowadays two efficient ways for transporting video: either the use of traditional protocols coupled with resource reservation protocols or the use more adapted new protocols. The first solution guarantees a constant quality of presentation but presents several drawbacks: resource reservation techniques are not yet globally implemented as well their use is expensive. We will show in the following that the second family of solutions although not able to guarantee a constant quality of service is able to provide an acceptable quality at a lower cost [15]. We thing that this second category of solutions is the most appropriate one in current widely used best effort distributed networks.

### 2.2 Related Work

Most research work in video transport use traditional transport protocols and do not consider the ability offered by video streams for relaxing order constraints as a mean for increasing performances.

Many solutions have been proposed for the transport of MPEG video streams. In [22] a forward error code variation technique named Priority Encoding Transmission (PET) is used. The performance gains entailed by this approach are notable. Nevertheless, PET increases codec complexity, CPU load and introduces a traffic overhead of about 25 %.

For minimizing the extent to which errors propagate within a MPEG sequence, [20] proposes a technique based on a less macro-block dependent decoding. Though this method introduces a light traffic overhead it suffers of being a “not compatible with the standard” approach. The MPEG2 standard [7] considers network issues by proposing two levels of priority for packets: a base level and enhanced level. The base level contains only low frequency DCT (Discrete Cosine Transform) coefficients and can be decoded independently, whereas the enhancement level carries high resolution data that is only useful if added back to the base layer. By separating the bit-stream, the bit rate per frame is slightly increased and the codec implementation is much more complex [3]. Furthermore, the base layer has to be transmitted over a reliable channel. Finally, [19] analyses the transport of several video standards in the framework of the ALF (Application Level Framing) concept. ALF proposes to customize transport functionality to the application needs by implementing this functionality at the application level. Such an approach enables to take into account the processing of out of order Application Data Unit (ADU). This contribution does not propose a real transport strategy for video streams but is one of the first works referencing the benefits obtained from exploiting ADUs’ order. In this context, [10] analyses the benefits of the ALF approach for the transport of M-JPEG video streams.

## 3 Order and Reliability laxity for MPEG Streams Transport

In the absence of QoS guaranteed paths (like in best-effort Internet), traditional protocols are not able to satisfy real time application requirements (particularly end-to-end delays or latency). In this context, it has been proved that partial order and reliability connections are able to provide more adequate services [16][23]. With respect to CO services, POC provide mainly two advantages: saving communication resources and reducing number of blocking times. Nevertheless, in order to obtain these benefits, real time applications must alleviate transport constraints such as reliability and order.

As far as MPEG video applications are concerned, they are time sensitive but they can also tolerate some losses and out of order delivery. MPEG video applications are then good candidates to exploit the advantages of the POC approach. In this section we determine the MPEG video elements that support out of order delivery or losses (if necessary), in order to maintain the continuity of the video presentation.

### 3.1 MPEG Compression Algorithm

The MPEG algorithm achieves compression in three basic phases:

(1) reduction of source resolution by filtering;
(2) removal of the spatial and temporal redundancies by applying subjectively adapted lossy compression techniques;
(3) compression of the resulting information by applying variable length coding techniques. Then, the bit-stream is assembled by combining both fixed length (the synchronization sequences and data headers) and variable length codes (compressed data).

The use of these techniques generates three types of pictures: Intra-picture (I), Predicted pictures (P) and Interpolated pictures (B – for Bi-directional prediction). Intra-coded pictures are coded by using spatial reduction redundancy techniques and their compression ratio is moderate. These pictures are coded (and then decoded) independently from any other pictures. Predicted pictures are coded with reference to a past picture (Intra or another Predicted) and in general will be used as a reference for future Predicted pictures. Finally, Bi-directional Pictures are encoded by interpolation from past and future pictures (see Figure 2). They are never used as references.

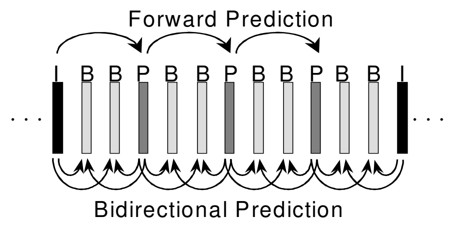


**Figure 2. Dependency Relationship between I, B, and P-pictures.**

The number of Predicted and Bi-directional pictures that are related to a single Intra-picture represents the *prediction scheme*. The prediction scheme can be changed according to application specific parameters such as random access and coding/decoding delays. The prediction scheme is characterized by two parameters: *M,* which specifies the distance (measured in number of pictures) between two consecutive Predicted pictures and *N,* which specifies the distance between two consecutive Intra-Pictures. For the sequence depicted in Figure 2, M = 3 and N = 12.

The MPEG video bit-stream structure is organized into a hierarchical structure (see Figure 3). A *sequence* is the top layer of the coding hierarchy and consists of a header and one or more *Groups Of Pictures* (*GOPs*). GOPs are usually (but no systematically) decoded by using previous GOP information. Two types of GOPs are identified: (1) open GOPs that must be decoded by using previous GOP information and (2) closed GOPs that are fully independent (a flag named closed_gop placed in the GOP header indicates this feature). A GOP consists of a header and a number of *pictures*. A *picture* corresponds to a single frame of motion video. Note that, because of picture dependencies, reference pictures (I and P) must be present at the decoder before their related bi-directional pictures (B); therefore, for the *display sequence* illustrated by Figure 2 the *transmission sequence* must be IPBBPBBPBBIBB. A Picture consists of a header and one or more *slices* which in turn consists of a header and one or more *macro-blocks*. Each of them consists of a header and six components named *blocks*, four blocks for luminance, the other for chrominance.

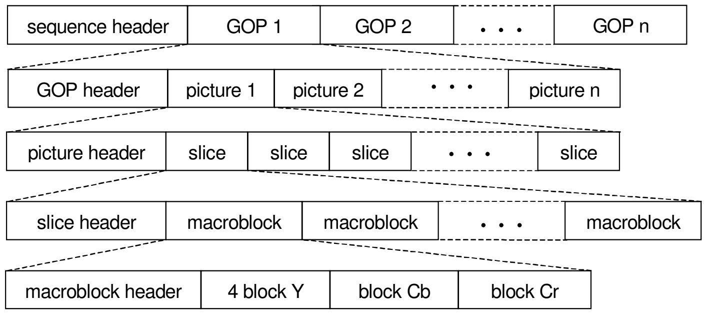


**Figure 3. The MPEG video data hierarchy.**

### 3.2 Admissible losses in MPEG Video Streams

When transmitted over a network, the MPEG video bit-stream structure enables the decoder to support missing elements. This capability offered by the MPEG coding/decoding process allows blocking delays to be avoided and then playback continuity to be maintained. Nevertheless, in order to maintain a good viewing quality, three aspects must be controlled: (1) the loss size (2) the loss type and (3) the loss ratio.

Firstly, in order to limit the impact of losses on viewing quality, the sender must use transmission units of a suitable size. For instance, if the provided service is not totally reliable, it is better to use small transmission units (such as slices or macro-blocks) than large ones (such as pictures or GOPS). Indeed, the bigger the lost is the worst the degradation is. We think slices are the optimal transmission unit because their size can be pre-determined by configuring the MPEG coder and they can be decoded in an independent way. In contrast macro-block have a very small fixed size (control information/payload ratio is not good) and can not be decoded in an independent way (their position, with respect to the slice header, is used during decoding).

Secondly, a data can either belong to a reference picture (I or P-Picture data) or not (B-Picture information). The loss of no referenced data will just produce damages in one picture whereas the loss of reference data will extend to past and future pictures because of the inter-picture relationship (Figure 4). In this context, reference data require more reliability than no reference data.

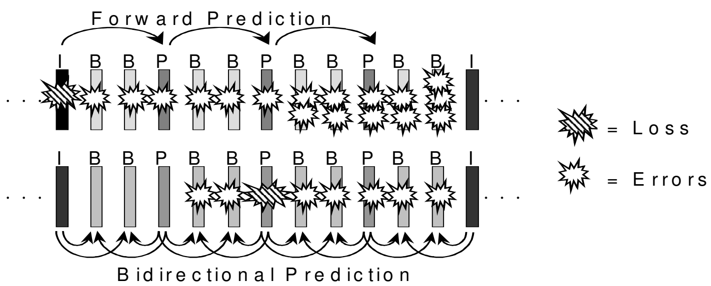


Figure 4. Influence of losses on final presentation.

Additionally, depending on the maximum compensation motion offset per frame, the damage produced by the loss of a reference data may expand through the successive frames of a GOP up to cover the entire screen (see Figure 4). The sender can reduce the maximum compensation motion offset for limiting the rate at which the damage can expand. Nevertheless, there is a need of a service offering some degree of reliability.

Thirdly, the loss ratio depends on the viewing quality the final user is ready to accept.

### 3.3 Admissible disorder in MPEG video Streams.

When the decoding of an ordered or unordered sequence of video elements gives the same viewing quality we say it admit unordered elements. The need of processing out of order elements arises from two reasons. Firstly, the network may reorder transmission units. Secondly, when a transmission unit is lost and then successfully retransmitted it will arrive out of order. In either case, unordered processing capabilities allow the decoder to be maintained busy face to lost or unordered transmission units.

The MPEG video standard does not formally specifies which elements of the video bit-stream can be processed out of order. However, due to their coding independence some of these elements enable out of order processing.

An analysis of the bit-stream layer hierarchy (see Figure 3) shows that both GOP level and picture level do not allow any out of order processing. GOPs must respect a fixed sequence of presentation, even if they are fully independent (§ 3.1) and they can be potentially decoded in any sequence. Due to the inter picture decoding dependencies, pictures must also be decoded in a strict order. On the other hand, macro-blocks and blocks have not enough information for being processed independently. In contrast, slices can be processed in any order without degrading the viewing quality (i.e. if disorder does not go beyond the picture's boundaries). When out of order slices exceed picture boundaries (we name this kind of disorder, *inter picture disorder*) the degradation is unavoidable. However, accepting slices with inter picture disorder allows the decoding to stop error extent (as we described above § 3.2).

Thus, by considering the previously evoked order and reliability issues we propose to use slices elements as transmission units.

It is worthy to consider that current implementations of MPEG video decoder are not able to decode out of order slices. Indeed, we have analyzed several software [27][30] as well as hardware [17] MPEG decoders. All these decoders exhibit problems with disordered slice sequences (they produce fuzzy pictures), even if they are only limited to the extent of a picture. Therefore, in order to take advantage of the MPEG video standard over a POC connection, we need to define a correct transport strategy (error recovery included) and create a decoder able to cope with out of order data.

## 4 A new Transport Service for MPEG video

This section introduces and motivates a transport service based on Partial Order Connections for MPEG video streams. This service aims to improve, compared to currently used transport service in best effort distributed environment, the quality of the MPEG video perceived by user. The proposed approach considers only point to point communication schemes. That is, the components involved in the proposed strategy are one sender, one receiver and the communication protocol. This transport service is defined in three steps: first transport service data units (TSDU) are defined, then we choose a formal model for reasoning precisely about the transport delivery of these TSDU and finally, we specify this new transport service by using this formal model.

### 4.1 Packetization of the MPEG Bit-stream

The goal of the packetization process is to create TSDUs that enable the receiver (in this case the MPEG decoder) to operate under network error condition (losses and disordered information). The definition of TSDU must consider two main aspects: the TSDU content and the TSDU size.

- TSDUs must contain entire semantic elements because we suppose the receiver neither reassemble nor reorder packets. Therefore, if a semantic element is spread out over several packet, then the decoding process would be impossible in cases of packet losses or disordering. That means the packetization process must ensure that semantic elements (such as slices or headers) are not spread out over several packets.
- The choice of the TSDU size is an essential aspect since it impacts on viewing quality (see § 3.2) as well on the fragmentation in lower layers, which can entail serious performance degradation [21]. A method to avoid lower layers' fragmentation is the creation of data packets no longer than the path's MTU (Maximum Transmission Unit) [26].

These considerations lead us to adopt slices or headers as the semantic elements that constitutes TPDUs. Slices are distinguished from headers because these two types of data require different transport QoS. Indeed slices can be lost or delivered out of order under certain limit (see § 3.3) while headers must conserve their transmission order and their loss entails serious video degradation. These considerations involve the use of two types of TPDU: Upper Layer Header packets (*ULH*) and Slice packets (*S*), where *ULH* packets can hold Sequence Headers (*SH*), GOP Headers (*GOPH*) or Frame Header (*FH*) whereas *S* packets can only contain frame slices (Figure 5).

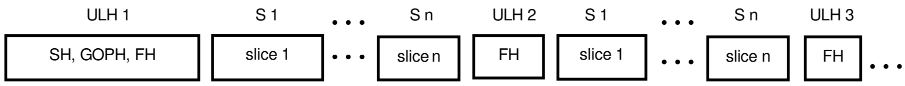


**Figure 5. Bitstream fragmentation and packetization.**

In some cases, several different headers are aggregated in a single *ULH* packet. That is essentially motivated by their small size and also because this aggregation represents a means to conserve their order. For instance, the aggregation of three headers (SH, GOPH and FH) only takes place at the beginning of a video sequence and a two headers aggregation (a GOPH and FH) is done at the beginning of the GOP. As far as *S* packets are concerned, they must contain only one slice which the size can be controlled by configuring the coder.

### 4.2 The TSPN model

The Hierarchical Time Stream Petri Net (HTSPN) model is a Petri net extension that allows one to easily and accurately reason about temporal and synchronization constraints in asynchronous systems. Using TSPN an action (i.e. a communication, presentation process) is modeled as an arc labeled by a 3-uple (arc composed by place F and transition t in Figure 6-a). This 3-uple, (x,n,y) in Figure 6-a, specifies respectively the minimal, nominal and maximum admissible duration of the considered process (i.e. all the durations which satisfy the semantics of the process). This temporal 3-uple allows the temporal non-determinism of processes in asynchronous distributed systems to be modeled. The dynamic behavior of a so modeled process is the following: a token arrival in place F at time $\tau$ models the start of the process: the related transition t can be fired and therefore the token withdrawn at any time in the time validity interval [$\tau$+x, $\tau$+y] (TVI). This synchronization scheme considers a basic case where one process is involved only; if we consider a synchronization scheme between several processes (Figure 6-b) then reasoning (and therefore modeling ) about synchronization between several processes which have an asynchronous behavior becomes more complex. Indeed, it can be shown that in the general case it is impossible to satisfy the temporal constraints of the all the processes. The TSPN model solves this problem by proposing a formal semantics of synchronization in asynchronous systems. The basic idea of this synchronization semantics consists in favoring a statically or dynamically defined process. Such an approach entails three fundamental strategies, that is: to favor the latest process (the "and" synchronization type), the earliest process (the "or" synchronization type), or a given pre-defined master process (the "master" synchronization type). The combination of these three basic synchronization types lead to a complete set of nine synchronization operators which define a semantics of synchronization in asynchronous systems. TSPN allow this synchronization semantics to be modeled by using typed transitions [32].

In the following the TSPN model is used for reasoning about and specifying a partial order and partial reliability service for the transport of MPEG video streams.

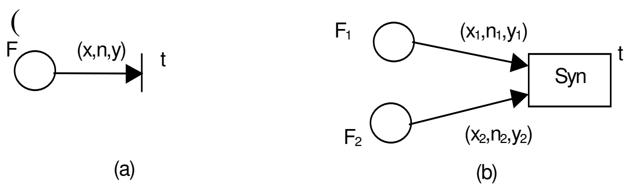


**Figure 6. Modeling of basic intra-stream (a) and inter-stream (b) synchronization schemes with HTSPN**

### 4.3 Definition of the partial order and the partial reliability transport strategy

With respect to continuous media applications, the limits of traditional protocols have been analyzed (§ 2.1), and the blocking times generation due to the inadequacy of the error recovery mechanisms has been identified as one of the most harmful problems.

This problem can be alleviated if applications reduce their transport constraints and protocols implements more adapted error recovery mechanisms. This principle represents the base of the transport strategy presented here. The idea is to specify a transport service that favors playback continuity by allowing transport constraints to be reduced. Of course, the transport constraints can only be reduced down to a minimal degree of viewing quality is maintained.

The transport strategy described in this section is divided in three parts: (1) definition of a transport QoS for each MPEG video component, (2) specification of the required multimedia QoS (partial order) and specification of the required monomedia QoS (reliability).

#### 4.3.1 QoS for MPEG video components (monomedia QoS).

In 3.2 and 3.3, the admissible losses and admissible out of order information that a MPEG video stream can support have been identified. From these considerations, three different (in terms of QoS) components have been identified: the synchronization information (requiring the most stringent QoS), reference pictures (requiring a intermediate QoS) and bidirectionally pictures (requiring a weak QoS). These components have the following transport needs:

- Synchronization information ( in ULH packets) requires a totally reliable and totally ordered service (§ 3.2 and 4.1). A total order is needed because these packets indicate the presentation picture order. A total reliability is required because the loss of one ULH packet makes impossible the decoding of the corresponding picture (contained in a set of S packets). That is because ULH packets contain the headers of the pictures. Moreover, in some cases the damage produced by the loss of one ULH can be more important since they may contain the GOP or sequence headers. This QoS will be referred as QoS 1.

- The slices (S packets) holding reference information (such as that held by Intra and Predicted Pictures) need a partial reliable and partial ordered service. Partial reliability can be accepted because MPEG authorizes the loss of some reference information. Remember that reference information is essential for decoding any downstream information. As far as partial order is concerned, slices have not order constraints inside a picture (§ 3.3) but the order between ULH packets and their corresponding S packets must be conserved. That is to say, the S packets belonging to a picture $P_i$ must be delivered after the related $ULH_i$ packet. Moreover, an additional control is required for I-Pictures and P-Pictures; the decoder requires I-Picture information before P-Picture because the decoding of P-Pictures depends on the decoding of I-Picture. Nevertheless, if a slice involving an I-Picture (or P-Picture) information arrives to late (after its presentation time) it is still accepted. It will not be used for presentation but it enables the decoder to limit extent of the degradation (see § 3.2 and 3.3). This QoS will be referred as QoS 2.

- Slices (S packets) holding no reference information (such as those held by Bi-directional Pictures) need a minimal reliable and a no ordered service. That is because the loss of a bi-directional information entails a very localized and limited viewing degradation (which is accepted for avoiding blocking times). As far as partial order is concerned there are no order constraints for the slices of a picture (§ 3.3). Nevertheless, the order between ULH packets and their corresponding S packets must be conserved. That to say, the S packets belonging to a picture $P_i$ must be delivered after the $ULH_i$ packet. For instance, slices belonging to the first B-Picture must be delivered after $ULH_2$ packet otherwise the decoding is not possible. That because $ULH_2$ contains the B-Picture Header of the first B-Picture (**Figure 7**). This QoS will be referred as QoS 3.

In the following, we will use the TSPN model as a modeling technique for specifying the multimedia QoS in terms of order. The monomedia QoS (reliability) is associated to each flow of packets, in the TSPN, as an attribute.

In order to illustrate how to specify the MM-POC service for a MPEG video sequence let us analyze an example. Consider a video sequence which the GOP structure corresponds to IBBPBB (this sequence has got a prediction scheme characterized by $M = 3$ and $N = 6$). Using the TSPN model, such video sequence can be modeled by the TSPN presented in **Figure 7**.

#### 4.3.2 Specification of the multimedia QoS (partial order)

Let us first consider the order dimension of the required service. The MM-POC service is based on the TSPN model. The formalism of this model allows an application to explicitly specify: (1) the set of objects and their relations (i.e. places and arcs) and (2) inter-object synchronization constraints (i.e. transitions firing rules). TSPN allow parallel and serial relations to be expressed. Parallel relations are used to express out of order delivery whereas serial relations are used to express total order. By combining serial and parallel relations, we can express any partial order sequence. Moreover, HTSPN have the expressive power of a Turing machine. Therefore, note that temporal constraints, which can be accurately expressed by using HTSPN, are for the moment out of the scope of the POC concept, and will not be considered. That means that TVI associated to modeled processes will be ignored here (i.e. they are all equal to $(0,*,+\infty)$, where * is for any instant in the TVI).

Therefore, the TSPN in **Figure 7**, specifies as follow the fundamental QoS parameters associated to the transport of a MPEG video stream:

The order component of QoS 1 (for ULH packets) is a totally ordered service. This QoS is specified by a serial sequence of places (**Figure 7**).

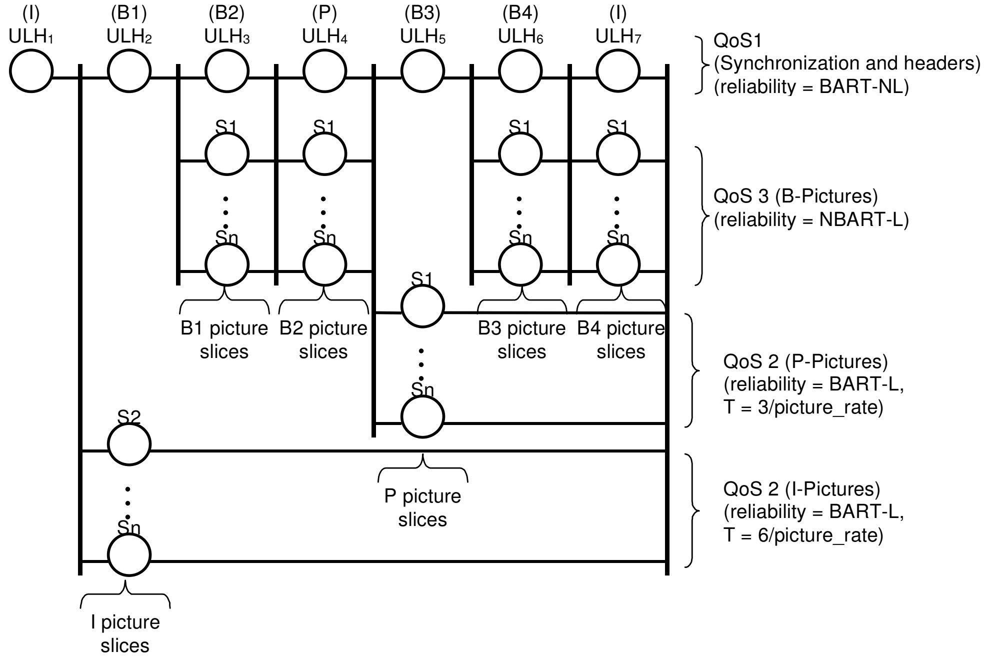


**Figure 7. HTSPN modeling of the POC transport service.**

The order component of QoS 2 is a partial order service (that concerns I and P reference information). In order to specify out of order delivery of S packets, parallel relations are used. On the other hand, we have previously indicated that before addressing a content data (S packet) to the MPEG decoder, the related header (ULH packet) must have been previously received and decoded. In order to express such relation we place each ULH place, just before the transition corresponding to the S place (**Figure 7**). Finally, I-Picture slices should usually be delivered before P-Pictures slices. Nevertheless, as slices of P-Pictures are authorized, for limiting error propagation, to arrive until the first header of the next GOP ($ULH_7$), I-Pictures slices are authorized to be delivered after a P-Picture slice. This is a generic scheme, also used for P-Pictures, which allows reference information to arrive late for reducing error extent (§ 3.2 and 3.3).

The order component of QoS 3 is a partial order service (that concerns B-Picture information). In this case, parallel relations are used for specifying the out of order delivery of S packets (inside a picture). On the other hand, we have previously indicated that before addressing a content data (S packet) to the MPEG decoder, the related header (ULH packet) must have been previously received and decoded. In order to express such relation we place each ULH place, just before the transition corresponding to the related S places (

**Figure 7**). For instance, S places representing Bi-directional information are always placed after a ULH place involving B-Picture header information.

### 4.3.3 Specification of the monomedia QoS (reliability)

As far as the reliability dimension of the order and reliability protocol space is concerned, the MM-POC can associate to each flow of packets a given transport reliability service. In the POC concept, there are three classes of reliability [1]. They are the BART-NL, BART-L and NBART-L (BART stands for Buffers, Acks, Retransmissions, Timeouts which are four elements employed to obtain reliability, and L (respectively NL) indicates that losses are permitted (respectively not allowed)). The BART-NL class does not accept losses and uses buffers and retransmission mechanisms to satisfy this constraint. This is a blocking service. The BART-L class does not accept losses during the object temporal duration (i.e. the loss is accepted if the temporal semantics of the object can be no more ensured). In other words, a total reliability is only ensured during a bounded time. This service is implemented by using a timer and communication buffers combined with retransmission mechanisms. This service is not blocking. The class NBART-L provides an unreliable service; consequently it uses neither communication buffers nor retransmission mechanisms. This is a non blocking service (such as UDP). Using these classes of reliability, the specification of the partial reliability for QoS 1, QoS 2 and QoS 3 is as follows:

First, we specify QoS 1, the reliability required to transport ULH packets. It must be of 100%; BART-NL class can provide it. Secondly, QoS 2 corresponds to the reliability required to transport S packets holding reference information must be 100%. Nevertheless, no blocking times are wanted. The BART-NL class can provide such service. The time T to be associated is N/2×picture_rate (where N is the distance between I-pictures and picture_rate is typically 25 Hz for PAL and 30 for NTSC) for packets S involving I-Picture information and M/picture_rate (where M is the distance between P-Pictures) for packets S involving P-Picture information. Finally, QoS 2 corresponds to the reliability required to transport S packets involving bi-directional information; it can be 0 %. NBAR-L class is, in this case, the most adequate service.

# 5 Performance evaluation

The POC strategy has been evaluated in comparison with TCP and UDP, mainly by considering the behavior of these protocols in presence of data losses and their impact on presentation quality of a video on-demand application. This section shows how and why the partial order strategy offers a good compromise between the loss of presentation quality induced by UDP with packet losses and the blocking duration entailed by the TCP approach.

## 5.1 Picture recovery

The POC transport strategy presented above is able to fix the error extent produced by the loss of slices involving reference information. In this section, we compare the gain of this strategy with respect to the results obtained by using an UDP-like protocol. In the following, the number of damaged pictures is calculated without error recovery mechanisms (the UDP-like protocol case) and then with error recovery mechanism (the POC protocol case). Then results are compared.

### 5.1.1 Number of damaged pictures without error recovery mechanisms

The dependency relationship induces a fixed number of errors, which can be easily determined from the GOP structure. For example, the loss of a slice of an I-Picture will damage the corresponding slices on all the following pictures of the related GOP. Moreover, this damage may extent to the first B-Pictures of the next GOP, if it is open (§ 3.1).

The number of reference pictures in a GOP $N_{ref}$ is simply given by the following expression (Where *M* and *N* are the prediction scheme parameters referred in section 3.1):

$$(1)\quad N_{ref} = \frac{N}{M}$$

The total number of damaged pictures $D(i)$ (due to the loss of a slice) in the i^th reference picture $P_i$ can be expressed by:

$$(2)\quad D(i) = [(N_{ref} - i).M - 1] + [(M - 1)_{i \neq 0}] + (M - 1)_{closed\_GOP = 0}$$

The first part of the expression gives the number of damaged pictures from $P_i$ to the end of the GOP. The second part express the number of damaged past Bi-directional pictures; that does not involve the first reference picture $P_0$ (I-Picture). Finally, the third part expresses the number of next GOP's Bi-directional pictures that are damaged by the dependency between GOPs (§ 3.1).

### 5.1.2 Number of damaged pictures in a POC approach.

By considering an error recovery mechanism based on retransmission, the communication system requires a time $T_r$ to recover a loss. As the POC service delivers data in a not blocking manner (it does not systematically stop during the error recover process), losses may only be perceived during a period of time $T_r$. The number of damaged pictures is then the number of displayed pictures during the period of time $T_r$. The number of displayed pictures during a period depends on the *picture_rate*. Thus, the number of damaged picture in a POC approach can be stated by the follow expression:

$$(3)\quad D'(i) = \min(D, T_r * picture\_rate)$$

$T_r$ is function of the round trip delay (RTT) and of the recovery mechanism, which is depending of the network characteristics. A typical RTT in France is 60ms. The *picture_rate* is either 30 for NTSC and 25 for PAL/SECAM.

### 5.1.3 Number of damaged pictures with and without error recovery mechanisms

By using a *picture_rate* = 25, a $T_r$ = 60ms and considering a GOP where *N*=12 and *M*=4, Figure 8 shows the number of damaged picture with and without recovery mechanism.

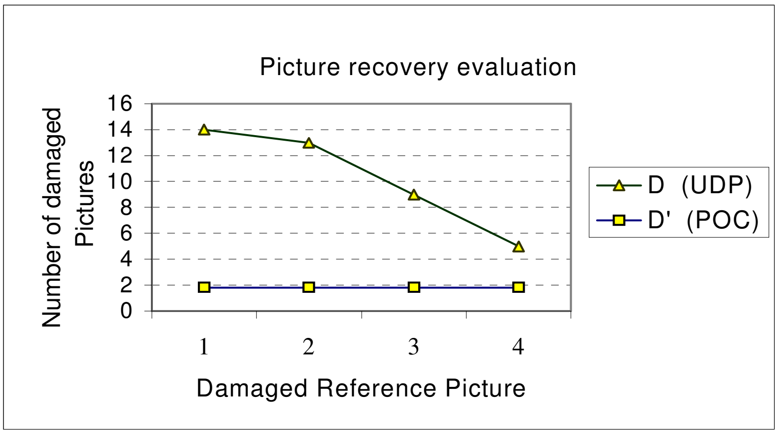


**Figure 8. Picture damage evaluation in case of slice loss.**

The above picture illustrates the impact of the recovery of some MPEG packets. Indeed, one error on a reference packet can affect an important number of pictures. By using a UDP connectionless service, important damages can occur if a lost packet involves a GOP header. In such a case, the whole pictures in the GOP are lost. Conversely, TCP-like strategy uses a recovery scheme for all the transmitted packets and it does ensure a total ordering of the transmitted packets. We study in the next section the effect of the TCP transport strategy onto the blocking time.

### 5.2 Blocking time

A POC protocol offers a service able to reduce (as specified by the application) the number of blocking times. In order to quantify the advantage of our transport strategy in term of blocking delay, the number of packet that could potentially introduce blocking times (with respect to a GOP) is compared to the total number of packets in a GOP.

Considering the packetization scheme presented in section 4.1, the total number of packets per GOP, $T$ , is given by the following expression (we consider a fixed number of slices per picture):

(4) $T = (N * slice\_per\_picture) + N$

The first term of the addition corresponds to the total number of slices of a GOP and the second term is for the number of ULH packets (see Figure 5). Considering the fact that only ULH packets can introduce blocking time with the POC transport strategy, the *percentage of blocking packet* that could introduce blocking delay $B_{\%}$ is given by:

(5) $B_{\%} = \frac{N}{T} * 100 = \frac{1}{slice\_per\_picture + 1} * 100$

Considering the QCIF format where each picture contains 15 slices, the percentage of blocking packets is 6,25%, value to be compared to 100% for a total order TCP service.

## 6 Implementation and performance results

Three transport services, i.e. connection less, connection oriented and partial order connection have been studied in the context of MPEG video transport. In order to consider the real effect of those transport strategies on a video presentation an evaluation platform has been implemented.

### 6.1 Evaluation platform

The functional architecture of the evaluation platform is based on four main subsystems: the sender, the receiver, the transport protocol implementing each of the transport services, and the network.

At sender side, the file handler extracts a MPEG video sequence from a file and creates the TSDUs according to the packetization scheme described in the above section (see § 4.1). Moreover, given that the MPEG video bit-stream does not include the necessary information to allow the recovery of disordered pictures (see § 3.3), the sender must add a sequence number to the transmitted packets. Nevertheless, the resulted bit-stream keeps a MPEG video coding compatibility, the ordering information being set into a reserved unused field of the MPEG stream format.

At receiver side, an ad-hoc MPEG decoder ensures the correct flow presentation. It as been specially designed to cope with disorder and unreliability according to a partial order transport strategy. However, it remains a fully compatible MPEG I decoder, the ordering information being set into a reserved unused field of the MPEG stream format. The implementation of our MPEG I decoder and player (mpeg_poc_play) is able to process MPEG I [6] bit-stream. This player is based on the MPEG II [7] player mpeg2play [27] of which are recovered DCT functions, movement compensation functions and some of the display routines for X11. In order to allow disordered information to be processed, references pictures that need to be presented but in which one or several slices are lacking (due to late or communication lost), are buffered. Downstream macro-blocks making reference to these missing slice(s) are also buffered. When a missing slice reaches the receiver, the decoder fixes the reference picture and all related downstream macro-blocks. These fixed pictures and blocks are then used to decode the incoming stream of pictures. We have measured that partial order support induces no more that 0.45% overhead with 5% of packets arriving out of their presentation time.

The transport service is supported by either UDP/IP, POC/UDP/IP or TCP/IP point to point sessions. Note that the goal of these experiences is not to compare the raw performances of these protocols, whose the services are very different, but rather to highlight the benefits of a POC approach, specifically in terms of service continuity and error recovery, with respect to the traditional approaches. On the other hand, in order to adapt the TCP/IP service to the needs of video transport, the `TCP_NODELAY` mode has been used. Therefore, the assembling of several TSDU into one TPDU (the classical behavior of TCP) has been avoided in order to reduce inter packet delay.

The network used is the best-effort Internet. The tests have been effectuated in the context of a Metropolitan Area Network situated in Toulouse city (France). The server was placed at LAAS/CNRS whereas the client was at ENSICA. Between these two places, there are approximately 10 kilometers and 5 IP nodes. The server runs over an Ultra Sparc 1 workstation and the client over an Ultra Sparc 2 workstation.

The obtained experimental results concerns two main aspects: (1) The impact on the viewing quality and (2) the continuity of service.

### 6.2 Impact on the viewing quality

Using the on-demand MPEG video application described in the previous section, the video transfer has been tested by using successively UDP, TCP and the proposed new transport service. The experiment has been achieved with a MPEG video sequence characterized by a prediction scheme with $M$=4 and $N$=12 and 15 slices per picture. The size of each TSPDU corresponds with the size of one picture slice. Considering pictures of 15 slices and a 25 frames per second rate, the inter TSDU period for emission is near to 2 ms. Figure 9 shows the three resulting sequences resulting with respectively UDP, TCP and POC.

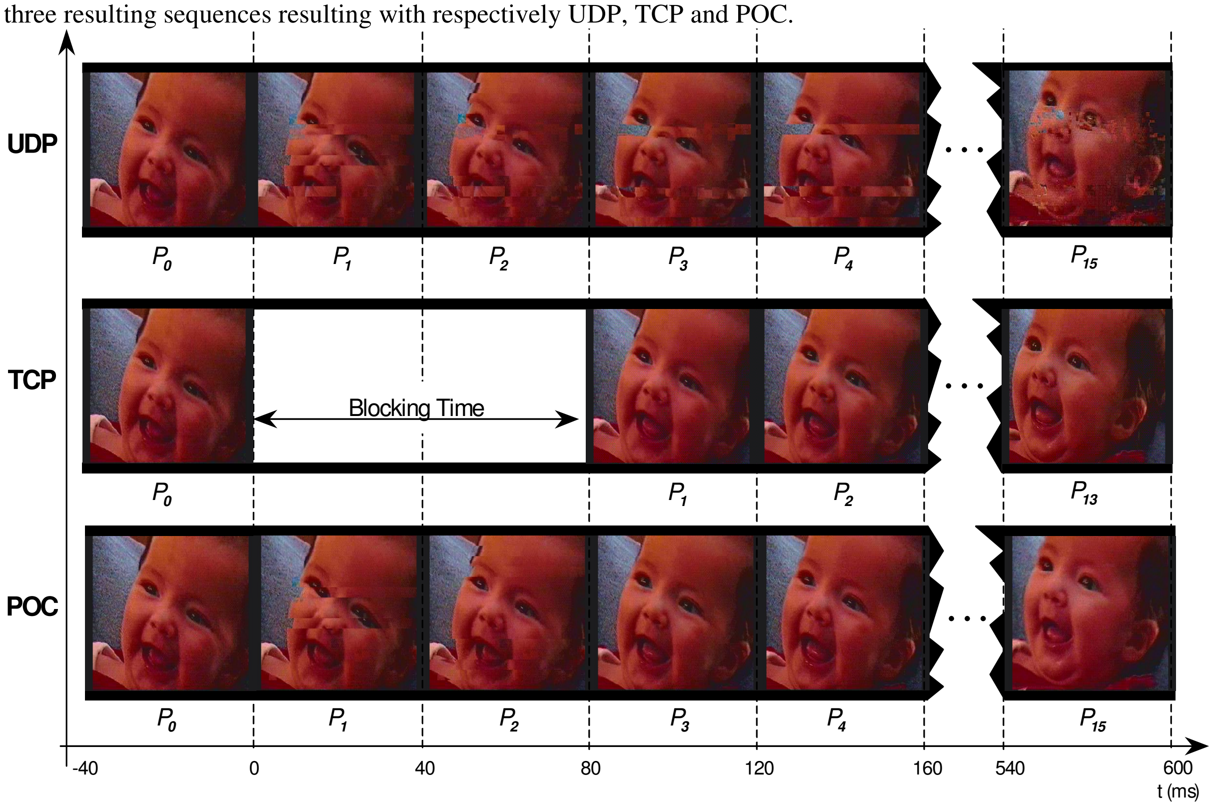

**Figure 9. Error recovery and playback continuity at final presentation level.**

Upon transmission of the shown sequence of pictures (which corresponds to one entire GOP), several slices have been lost: slices 6, 12 and 15 from picture $P_1$ (I frame) , slices 6 and 9 from picture $P_2$ (B frame) and slices 2, 4, 11,12 and 14 from picture $P_3$ (B frame). That is to say 5.56 % of packet lost in a GOP.

The slice loss with the UDP approach damages 15 pictures from the I picture of current GOP to the third B picture of the next GOP because current GOP is open (see § 3.1). The TCP approach does not alter any of the presented pictures. Nevertheless, the recovery scheme makes the stream to be paused during the time necessary to retransmit the lost information, introducing a discontinuity in the video stream. In the presented experiment, the recovery delay was approximately the current RTT (80 ms) which is the shortest duration that can be entailed by a recovery mechanism for data loss processing. The resulting blocking time corresponds to a two picture duration. In contrast, the POC approach can be considered as a good compromise between the two other approaches. Indeed, the damage caused by the loss is limited and there is no blocking time. As it can be seen in the last video sequence, the only pictures that are damaged by losses are $P_1$ and $P_2$. Picture $P_3$ and the following ones are fixed in a time corresponding to the recovery delay necessary to get the lacking reference slices (§ 5.1.2).

As we can see, the POC protocol offers an optimal error recovery mechanism. Actually, only TPDUs whose recovery is necessary (such as those involving Upper Layer Headers) can potentially originate a blocking time. In order to illustrate such a behavior, the video sequence presented on Figure 9 is used. We represent the traces of the incoming packets during the reception of such a video sequence (Figure 10).

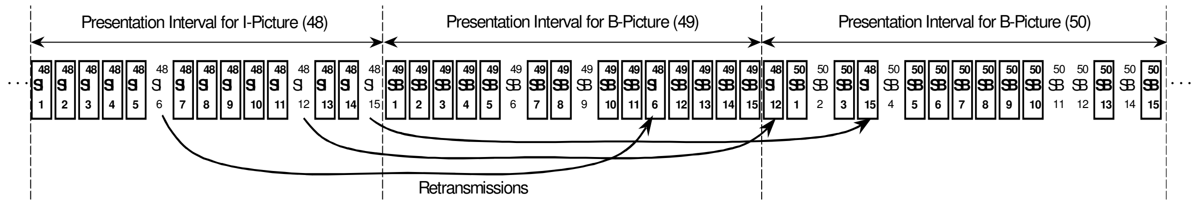


**Figure 10. Optimized error recovery**

The notation $ST_n^p$ represents the slice $n$ from the frame $p$ whose the type is $T$ (i.e. I, B or P). When this notation is inside a box it means that the slice has been received, else it has been lost. In this last case, an arrow represents the new position induced by the error recovery mechanism. For instance, slice $SI_6^{48}$ has been lost and the error recovery mechanism delays the delivery such that it arrives after its presentation time. Note that only losses corresponding to I type slices are successfully recovered. That is because their interval of validity is longest than the RTT. In contrast, B type slices always arrive after their corresponding interval of validity, they are then rejected by the POC protocol. Note that the absence of time references leads the POC protocol to try unsuccessful retransmissions due to an excessive RTT. Actually, during the transmission of 960 frames, 112 slices of type B have been lost and none was successfully recovered. Therefore, the association of a time reference (such as the TSPN TVI, § 4.2) and the RTT could allow the sender to decide the retransmission of one lost TPDU [29].

## 6.3 Impact on service continuity

In order to estimate the impact of a POC service on service continuity, the TCP service has been used as reference. As we have seen above, the loss of TPDUs generates blocking times, which in turn generates a viewing degradation. Figure 11 illustrates the distribution frequency of inter packet time of a POC service versus a TCP service, both under the same network conditions.

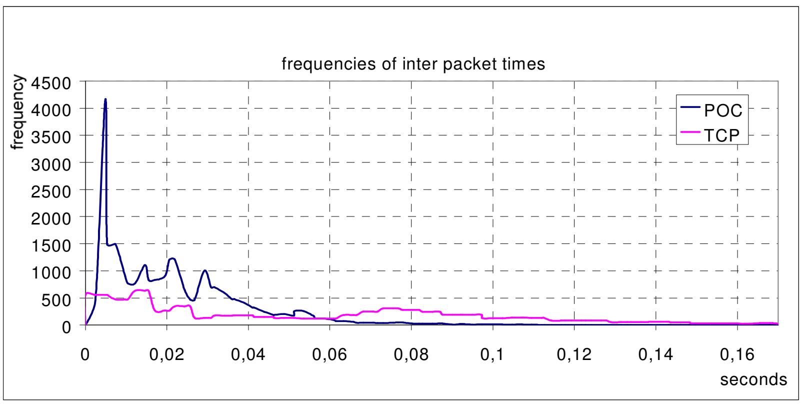


**Figure 11. frequency distribution of inter packet time**

For a TCP connection, the frequencies of inter packet time are more spread than those of a POC connection (Actually the mean inter packet time for TCP is 0.06 s and for POC it is 0.02 s). For TCP, frequencies are placed from 0 to 200 ms whereas frequencies for POC are concentrated in an interval that goes from 0 to 30ms.

# 7 Concluding remarks

We have presented a transport strategy for MPEG video that follows the POC approach. It has been shown that by accepting partial order sequences, transport service performances are greatly improved compared to traditional approaches. At the user level, advantages of the proposed approach can be objectively perceived, especially in terms of error propagation and playback continuity. Moreover, we have shown how a formal modeling technique can help to define precisely and unambiguously this new transport service.

It is important to note that the use of POC services is reserved to certain type of applications. In case of video client server applications, the possibility of using a POC transport strategy is offered by the semantic structure of the coded information, that allows (1) out of order processing to be achieved and (2) the decoder to tolerate information losses. Moreover, although the bit-stream of the most current video standard allows to implement POC strategies, the available decoders are not able to cope with POC principles. As far as MPEG video is concerned, we had to implement specific MPEG video decoder able to cope with out of order information.

For testing the proposed transport service under network conditions we have implemented a client-server system for video transport by using the services of TCP, UDP and POC. The performances of this system have been tested under best-effort network conditions to demonstrate the quality and performance gains that can be obtained compared with classical TCP or UDP approaches. We have demonstrated that a POC service fills not only the conceptual gap between TCP and UDP but also offers real performances improvements for the transport of multimedia streams such as MPEG video.

As far as future work is concerned, several parameters that must be accurately determined (since they influence the quality of final presentation). These parameters relate to the video structure and to the network. A correct choice of these parameters may improve the performances of our strategy. The parameters related to the video structure include the GOP depth (the number of references pictures contained in a GOP), the prediction scheme, the picture structure (i.e. its number of slices) and even the type of the scene. The parameters related with the network conditions relate essentially to loss rate and RTT. Here the question is: which is the best trade-off between the parameter of the MPEG video and the network conditions expressed in terms of loss rate and RTT. It is also interesting to accurately evaluate the impact of this strategy on network resources such as buffer occupancy, end to end delays, rate increase (some results can be found in [16]). Moreover, it is important to study how resource reservation techniques can be used in order to guarantee delivery of privileged streams (such as the header stream).

Finally, we plan to introduce at the transport level temporal constraints derived from application requirements [29]. Theoretical and experimental studies must be done for analyzing the impact of this new time dimension when introduced in the partial order-reliability protocol space (as in § 6.2).

# 8 Acknowledgements

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